



# UNITED STATES PATENT AND TRADEMARK OFFICE

*M/L*  
UNITED STATES DEPARTMENT OF COMMERCE  
United States Patent and Trademark Office  
Address: COMMISSIONER FOR PATENTS  
P.O. Box 1450  
Alexandria, Virginia 22313-1450  
[www.uspto.gov](http://www.uspto.gov)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/019,617	05/28/2002	Ravi Chandran	12447US03	6430
7590 McAndrews Held & Malloy 34th Floor 500 West Madison Street Chicago, IL 60661		02/05/2007	EXAMINER WOZNIAK, JAMES S	
			ART UNIT 2626	PAPER NUMBER
SHORTENED STATUTORY PERIOD OF RESPONSE		MAIL DATE	DELIVERY MODE	
3 MONTHS		02/05/2007	PAPER	

Please find below and/or attached an Office communication concerning this application or proceeding.

If NO period for reply is specified above, the maximum statutory period will apply and will expire 6 MONTHS from the mailing date of this communication.

<b>Office Action Summary</b>	Application No.	Applicant(s)
	10/019,617	CHANDRAN ET AL.
	Examiner James S. Wozniak	Art Unit 2626

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --  
**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

#### Status

- 1) Responsive to communication(s) filed on 08 November 2006.  
 2a) This action is FINAL.                    2b) This action is non-final.  
 3) Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

#### Disposition of Claims

- 4) Claim(s) 1,3-34 and 36-66 is/are pending in the application.  
 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.  
 5) Claim(s) \_\_\_\_\_ is/are allowed.  
 6) Claim(s) 1,3-34 and 36-66 is/are rejected.  
 7) Claim(s) \_\_\_\_\_ is/are objected to.  
 8) Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

#### Application Papers

- 9) The specification is objected to by the Examiner.  
 10) The drawing(s) filed on 28 May 2002 is/are: a) accepted or b) objected to by the Examiner.  
 Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).  
 Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).  
 11) The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

#### Priority under 35 U.S.C. § 119

- 12) Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).  
 a) All    b) Some \* c) None of:  
 1. Certified copies of the priority documents have been received.  
 2. Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.  
 3. Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

\* See the attached detailed Office action for a list of the certified copies not received.

#### Attachment(s)

- |   |   |
|---|---|
| 1) <input checked="" type="checkbox"/> Notice of References Cited (PTO-892)   | 4) <input type="checkbox"/> Interview Summary (PTO-413)           |
| 2) <input type="checkbox"/> Notice of Draftsperson's Patent Drawing Review (PTO-948)                                  | Paper No(s)/Mail Date. _____                                      |
| 3) <input checked="" type="checkbox"/> Information Disclosure Statement(s) (PTO/SB/08)<br>Paper No(s)/Mail Date _____ | 5) <input type="checkbox"/> Notice of Informal Patent Application |
|   | 6) <input type="checkbox"/> Other: _____                          |

## **DETAILED ACTION**

### *Response to Amendment*

1. In response to the office action from 6/12/2006, the applicant has submitted an amendment, filed 11/8/2006, amending claims 1, 3, 14, 26-29, 34, 36, 47, and 59-62, while arguing to traverse the art rejection based on the limitation regarding the decoding of speech and noise parameters at a noise controlling apparatus (*Amendment, Page 25*). Applicant's arguments have been fully considered, however the previous rejection is maintained due to the reasons listed below in the response to arguments.
2. In response to the amendment of claims 3, 14, 26-28, 36, 47, and 59-61, the examiner has withdrawn the previous claim objections drawn towards minor informalities.
3. In response to the amendment of claims 1, 29, and 62, the examiner has withdrawn the previous 35 U.S.C. 112, first paragraph rejection directed towards single means claims.

### *Response to Arguments*

4. Applicant's arguments have been fully considered but they are not persuasive for the following reasons:

In response to the applicants' arguments that amended claim 62 overcomes the 35 U.S.C. 101 rejection (*Amendment, Page 24*), the examiner notes that controlling the noise characteristic of a compressed digitized audio signal still constitutes non-statutory subject matter because the claim is still directed to abstract digital data (*0's and 1's*) and not a tangible audio output. In other words, the final result must be useful, tangible, and concrete, whereas manipulated digital bits are still abstract (*see Interim Guidelines for examination of Patent Applications for Patent Subject Matter Eligibility, Pages 37-39*). Thus, the 35 U.S.C. 101 rejection of Claims 62-66 is maintained.

With regard to independent claims 1, 29, 34, and 62, the applicant argues that Yue et al (*U.S. Patent: 6,026,356*) fails to teach any decompression of the speech signal at a noise controlling device, whereas the presently amended invention recites that decoding is performed at the noise managing apparatus (*Amendment, Page 25*). In response, the examiner notes that Yue does, in fact, teach that partial decoding is performed to extract LPC parameters at a noise controlling apparatus upon detection of noisy LPC parameters (extraction and synthesis processing Col. 5, Line 66- Col. 6, Line 44). This partial decoding process yields LPCs resulting in clean speech and resulting in noisy speech (*decoded preceding speech frames and current noisy frames, Col. 6, Lines 5-53*), which are then utilized in creating a noise-conditioned substitute frame that replaces the current LPC parameter frame (*Col. 6, Line 29- Col. 8, Line 11*). Thus, since Yue teaches a processor that is responsive to noisy LPC parameters to achieve an adjusted LPC substitute frame that is generated using decoded speech and noise LPC parameters, Claims 1, 29, 34, and 62 remain rejected.

The dependent claims are argued as further limiting rejected independent claims (*Amendment, Pages 25-27*), and thus, also remain rejected.

***Claim Rejections - 35 USC § 101***

5. 35 U.S.C. 101 reads as follows:

Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

6. **Claims 62-66** are rejected under 35 U.S.C. 101 because the claimed invention is directed to non-statutory subject matter.

As per the MPEP (2106 [R-3], IV):

In practical terms, claims define nonstatutory processes if they:

- ... consist solely of mathematical operations without some claimed practical application (i.e., executing a “mathematical algorithm”); or
- ... simply manipulate abstract ideas, e.g., a bid (*Schrader*, 22 F.3d at 293-94, 30 USPQ2d at 1458-59) or a bubble hierarchy (*Warmerdam*, 33 F.3d at 1360, 31 USPQ2d at 1759), without some claimed practical application.

In the particular case of Claim 62, the claimed subject matter is directed towards a method comprising “adjusting first bits and second bits,” which is merely a manipulation of abstract data in a processing device that does not, in itself, produce a useful, concrete, and tangible result.

Dependent claims 63-66 do not remedy the non-statutory subject matter issue noted above with respect to claim 62, and therefore, are also rejected under 35 U.S.C. 101, as being directed towards non-statutory subject matter.

***Claim Rejections - 35 USC § 102***

7. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

8. **Claims 1, 3, 13-15, 16-18, 20-21, 25-31, 34, 36, 46-48, 49-51, 53-54, and 58-64** are rejected under 35 U.S.C. 102(e) as being anticipated by Yue et al (*U.S. Patent: 6,026,356*).

With respect to **Claims 1 and 34**, Yue discloses:

A receiver receiving digital signals, wherein the digital signals use a compression code comprising a predetermined plurality of parameters including a first parameter, wherein the parameters represent an audio signal, the audio signal has a plurality of characteristics including a noise characteristic and is decodable by a plurality of decoding steps (*receiving coded speech frames, which include noise, Col. 5, Lines 31-65*);

A processor responsive to the compression code of the digital signals to read at least a first parameter, wherein the reading includes partially decoding the first parameter (*analyzing a compressed speech data frame and parameter extraction, Col. 5, Lines 50-Col. 6, Line 53*);

Responsive to the compression code and the first parameter, generating an adjusted first parameter and replacing the first parameter with the adjusted first parameter (*generating and substituting new LPC coefficients, Col. 5, Line 66- Col. 8, Line 11*).

The processor performs the plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoded signals resulting in an estimated clean speech signal, and wherein said processor responds at least to said first decoder signals and said second decider signals and said first parameter to generate said adjusted first parameter (*generating a correction factor based on new clean speech and original noisy LPCs, Col. 7, Line 29- Col. 8, Line 11*).

With respect to **Claims 3 and 36**, Yue discloses:

The first parameter comprises codebook gain, and wherein the processor modifies the codebook gain to modify the codebook vector contribution to the noise characteristic (*codebook gain and gain correction, Col. 3, Lines 47-67; and Col. 4, Lines 47-67*).

With respect to **Claims 13 and 46**, Yue discloses:

The plurality of parameters comprises pitch gain wherein the plurality of parameters further comprises a codebook gain, wherein said processor comprises a pitch synthesis filter, wherein said processor performs said plurality of decoding steps to generate a first vector, wherein said processor scales said first vector by said codebook gain to generate a scaled codebook vector, wherein said processor filters said scaled codebook vector through said pitch synthesis filter to generate a second vector, wherein said processor generates a power signal representing the power of said second vector, wherein said processor is responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain (*excitation parameters comprising codebook gain and LPC coefficients, which are processed using a synthesis filter to obtain information*

*representing power and modified according to a correction factor, Col. 3, Line 36- Col. 4, Line 15; and Col. 6, Line 30- Col. 8, Line 11).*

With respect to **Claims 14 and 47**, Yue further discloses the use of LPC excitation parameters corresponding to entries in a codebook (*Col. 3, Line 36- Col. 4, Line 14*).

With respect to **Claims 15 and 48**, Yue discloses:

The first parameter comprises a codebook vector comprising pulses using variable sets of amplitudes, wherein said processor analyzes said sets to identify the powers of said noise characteristic represented by said sets, wherein said processor identifies a first set representing a power less than the power represented by said sets other than said first set, and wherein said processor adjusts said pulses according to said first set to generate said adjusted parameter (*detecting noise coefficient sets which have a lower power than speech coefficients and using the coefficients in adjusting LPC coefficients, Col. 5, Line 66- Col. 8, Line 11; and Col. 3, Line 36- Col. 4, Line 14*).

With respect to **Claims 16 and 49**, Yue discloses:

The plurality of decoding steps further comprises at least one decoding step that does not substantially affect the management of the noise characteristic and wherein the processor avoids performing the at least one decoding step (*avoiding decompression processing, Col. 8, Lines 36-46*).

With respect to **Claims 17 and 50**, Yue discloses:

The one decoding step comprises post-filtering (*avoiding synthesis processing, Col. 5, Lines 50-65*).

With respect to **Claims 18 and 51**, Yue discloses:

The compression comprises a linear predictive code (*Col. 4, Lines 9-15*).

With respect to **Claims 20 and 53**, Yue discloses:

The compression code comprises code-excited linear prediction code (*CELP, Col. 1, Lines 18-32*).

With respect to **Claims 21 and 54**, Yue discloses:

The first parameter is a quantized first parameter and wherein the processor generates the adjusted parameter in part by quantizing the adjusted first parameter before replacing the first parameter with the adjusted first parameter (*quantizing correction bits before re-insertion into a data frame, Col. 4, Lines 47-67*).

With respect to **Claims 25 and 58**, Yue discloses:

The processor is responsive to the compression code to perform at least one of a plurality of the decoding steps to generate decoded signals and wherein the processor is responsive to the decoded signals and the first parameter to generate the adjusted first parameter (*extracting LPC coefficients from a compressed speech signal for noise modification, Col. 5, Line 66- Col. 8, Line 11; and Fig. 4*).

With respect to **Claims 26 and 59**, Yue discloses:

The first parameter is selected from the group consisting of: codebook vector, codebook gain, pitch gain, and LPC coefficient representations, including line spectral frequencies and log area ratios (*Col. 3, Line 36- Col. 4, Line 14*).

With respect to **Claims 27 and 60**, Yue discloses:

The first parameter comprises a representation of LPC coefficients, wherein the processor is responsive to the compression code and the representation to determine the spectral regions

affected by noise and to generate the adjusted first parameter to manage the noise characteristic in those regions and wherein the adjusted first parameter comprises an adjusted representation of LPC coefficients (*noise adjustment of LPC coefficients utilizing speech detection, Col. 5, Line 66- Col. 8, Line 11*).

With respect to **Claims 28 and 61**, Yue discloses:

The representation of LPC coefficients is selected from the group consisting of line spectral frequencies and log area ratios (*Col. 4, Lines 9-15*).

With respect to **Claims 29 and 62**, Yue discloses:

A receiver receiving digital signals, wherein the digital signals use a compression code comprising a predetermined plurality of parameters including a first parameter, wherein the parameters represent an audio signal, the audio signal has a plurality of characteristics including a noise characteristic and is decodable by a plurality of decoding steps (*receiving coded speech frames, which include noise, Col. 5, Lines 31-65*);

A processor responsive to the second bits to adjust the first bits and second bits, whereby the noise characteristic in the digital signals is controlled (*replacing LPC parameters for noise conditioning in response to analyzing an excitation bit segment at a speech detector, Col. 4, Lines 1-7; and Col. 5, Line 66- Col. 8, Line 11*).

The processor performs the plurality of decoding steps by performing first decoding steps to generate first decoder signals resulting in a noisy speech signal and second decoding steps to generate second decoded signals resulting in an estimated clean speech signal, and wherein said processor responds at least to said first decoder signals and said second decider signals and said

first parameter to generate said adjusted first parameter (*generating a correction factor based on new clean speech and original noisy LPCs, Col. 7, Line 29- Col. 8, Line 11*).

With respect to **Claims 30 and 63**, Yue discloses:

The linear code comprises pulse code modulation (PCM) code (*digitized audio samples obtained through PCM, Col. 3, Lines 23-35; and Col. 2, Lines 57-61*).

With respect to **Claims 31 and 64**, Yue discloses:

The compression code samples conform to the tandem-free operation of the global system for mobile communications standard (*Col. 1, Lines 18-32 and Col. 3, Lines 9-19*).

#### ***Claim Rejections - 35 USC § 103***

9. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

10. **Claims 4-5, 22-24, 37-38, and 55-57** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Swaminathan (*U.S. Patent: 5,495,555*).

With respect to **Claims 4 and 37**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest substituting speech coefficients based upon pitch gain, codebook gain, and signal to noise ratio (SNR), however Swaminathan discloses a means for

selecting an optimal codebook gain based upon a pitch gain, codebook gain, and a SNR (*Col. 12, Lines 40-67*).

Yue and Swaminathan are analogous art because they are from a similar field of endeavor in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the optimal codebook gain selection means disclosed by Swaminathan in order to achieve effective speech coding processing for voiced to unvoiced transitions (*Swaminathan, Col. 3, Lines 31-41*).

With respect to **Claims 5 and 38**, Swaminathan further discloses:

The signal to noise ratio comprises a ratio involving noisy signal power and noise power of the audio signal (*SNR, which is a ratio of signal to noise power, Col. 12, Lines 40-67*).

With respect to **Claims 22 and 55**, Yue teaches the system for reducing noise in compressed speech frames by substituting adjusted speech coefficients, as applied to Claims 1 and 34, while Swaminathan discloses subframe-based processing (*Col. 2, Line 60- Col. 7, Line 41*).

With respect to **Claims 23 and 56**, Yue further discloses the frame-by-frame noise processing method and system as shown in Fig. 4.

**Claims 24 and 57** contain subject matter similar to Claims 22 and 55, and thus, are rejected for the same reasons. Also, Yue further discloses generating substitute LPC coefficients based on past speech frame coefficients (*Col. 6, Lines 1-28*).

11. **Claims 6, 9, 39, and 42** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Oshikiri et al (*U.S. Patent: 5,878,387*).

With respect to **Claims 6 and 39**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest the use of a codebook gain and pitch gain at a buffer as recited in claims 6 and 39, however Oshikiri discloses the use of such gain factors (*codebook gain and optimal pitch gain selection, Col. 11, Lines 1-8; and Col. 12, Line 1- Col. 13, Line 11*).

Yue and Oshikiri are analogous art because they are from a similar field of endeavor in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the use of a codebook gain and pitch gain at a buffer as disclosed by Oshikiri in order to provide a means for obtaining sufficient pitch information for high quality voice reproduction at a decoder (*Oshikiri, Col. 4, Lines 12-17*).

With respect to **Claims 9 and 42**, Oshikiri further discloses:

The first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein the processor performs the plurality of decoding steps to generate a codebook vector, wherein said processor scales said codebook vector by said codebook gain to generate a scaled codebook vector, wherein said processor generates a power signal representing the power of said scaled codebook vector, wherein said processor is responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain (*multiplying a vector by an optimal codebook gain to determine a power signal for error determination which is used to calculate an optimal pitch gain, Col. 7, Line 33- Col. 8, Line 21; and Col. 8, Lines 46-65*).

12. **Claims 7-8 and 40-41** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Ertem et al (*U.S. Patent: 6,453,289*).

With respect to **Claims 7 and 40**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest adjusting speech coefficients based upon pitch gain and signal to noise ratio (SNR), however Ertem discloses a means for selecting a pitch gain correction factor based upon a pitch gain and an estimated SNR (*Col. 10, Lines 19-45*).

Yue and Ertem are analogous art because they are from a similar field of endeavor in noise reduction in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the gain correction factor selection means disclosed by Ertem in order to achieve reliable noise estimation for noise reduction processing (*Ertem, Col. 1, Lines 38-44*).

With respect to **Claims 8 and 41**, Ertem further discloses:

The signal to noise ratio comprises a ratio involving noisy signal power and noise power of the audio signal (*SNR comprising speech and noise power levels, Col. 10, Lines 19-45*).

13. **Claims 10-12, 19, 43-45, and 52** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Chen (*U.S. Patent: 5,615,298*).

With respect to **Claims 10 and 43**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest the use of a pitch gain at a buffer as recited in claims 10

and 43, however Chen discloses a process for determining pitch weighting for a first lag (*Col. 28, Line 11- Col. 29, Line 18*).

Yue and Chen are analogous art because they are from a similar field of endeavor in noise reduction in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the process for determining pitch weighting for a first lag as taught by Chen in order to achieve pitch processing which ensures that voiced regions do not get amplified relative to unvoiced regions (*Chen, Col. 29, Lines 11-14*).

With respect to **Claims 11 and 44**, Chen discloses a process for determining pitch weighting for a second lag (*Col. 28, Line 11- Col. 29, Line 18*).

With respect to **Claims 12 and 45**, Chen discloses a long-term predictor buffer utilized for the first and second pitch lags (*Col. 28, Line 11- Col. 29, Line 18*).

With respect to **Claims 19 and 52**, Chen further discloses the use of a long-term predictor code (*Col. 11, Lines 26-33*).

14. **Claims 32-33 and 65-66** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Navaro et al (*U.S. Patent: 6,108,560*).

With respect to **Claims 32-33 and 65-66**, Yue discloses the system for replacing speech parameters, as applied to Claims 29 and 62. Yue does not teach system implementation in TFO GSM format that comprises 2 LSBs and 6 MSBs of PCM speech data, however Navaro teaches speech coding implemented in such a format (*Col. 6, Lines 11-29*).

Yue and Navaro are analogous art because they are from a similar field of endeavor in speech coding systems. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the speech coding implementation in a GSM system format as taught by Navaro in order to achieve high quality speech coding in a mobile environment (*Navaro, Col. 1, Line 11- Col. 2, Line 11*).

***Conclusion***

15. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

16. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

Artsuka et al (*U.S. Patent: 5,185,848*)- discloses noise reduction at a coded speech receiver.

Chan et al (*U.S. Patent: 5,771,486*)- utilizes a clean speech signal and a noise level for determining a noise suppression characteristic.

Thyssen et al (*U.S. Patent: 6,240,386*)- discloses noise detection and compensation at a decoder to identify the existence of noise in a speech signal and to determine if the noise should be compensated during the processing of the speech signal.

Barker et al ("Decoding Speech in the Presence of Other Sound Sources," 2000)- discloses a method for decoding speech in noise.

17. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (571) 272-7632. The examiner can normally be reached on M-Th, 7:30-5:00, F, 7:30-4, Off Alternate Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David Hudspeth can be reached at (571) 272-7843. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Art Unit: 2626

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see <http://pair-direct.uspto.gov>. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

James S. Wozniak  
1/18/2007



DAVID HUDSPETH  
SUPERVISORY PATENT EXAMINER  
TECHNOLOGY CENTER 2600